



from INSPIREDACOUSTICS.

User Manual

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1 Welcome



Welcome to the INSP:IR Impulse Library Vienna Edition and congratulations for your purchase!

INSP:IR Impulse Library (ILL) Vienna Edition is a collection of sampled reverberation (impulse responses) from renowned venues designed to operate within the Vienna Symphonic Library (VSL) Vienna Suite Convolution Reverb engine. The *INSP:IR Impulse Library (ILL) Vienna Edition* is the essential inspiring tool you need to get the most out of Vienna Suite's Convolution Reverb.

1.1 Highlights

- 10 different halls ranging from intimate halls to large basilicas
- 140 dB dynamic range (full 24-bit)
- 174 stereo impulses
- 56 true stereo impulses
- and more

2 Installation

2.1 Prerequisites

The ILL Vienna Edition requires a fully licensed installed version of *Vienna Suite Convolution Reverb*. For detailed information on the hardware and software requirements of the Vienna Suite, please consult the VSL website.

Web: The Vienna Suite Convolution Reverb is part of the Vienna Suite which can be obtained from the VSL website: <http://vsl.co.at/en/211/497/1686/454/115.htm>

2.2 Installation

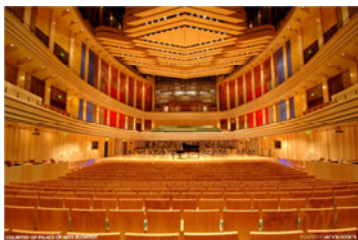
Installation of the ILL Vienna Edition is easy, just run the installer that you received during your purchase and the software will automatically install. Make sure you restart the Vienna Suite Convolution Reverb to make the new presets appear.

3 Halls and preset naming convention

There are all together 10 halls in the IIL Vienna Edition, and the built-in presets are provided so that you can use them as stereo, true stereo and quad setups flexibly.

The naming of the presets are very simple and on the bottom left side of the VSL Convolution Reverb there are explanations where the sound sources and microphones were located.

3.1 Halls in the IIL Vienna Edition



Béla Bartók National Concert Hall

The Bartók National Concert Hall is located in the heart of the building of the Palace of Arts, Budapest, Hungary. It was built in 2005, and it was awarded with the Prix d'Excellence of FIABCI in 2006, the international 'Oscar Award' for construction and real estate development.

The acoustics of the concert hall is adjustable, with the resonant chamber system placed next to the side walls and the podium, the large movable doors serving to modify the size of the concert hall. Using an apparatus found in the auditorium of the Bartók National Concert Hall called 'the canopy', the height can be regulated in a range of 18 metres by computer control – which also influences the acoustics of the hall. The canopy weighs 45 tons, and is suspended on 40 steel cables. The hall is 25 m high, 25 m wide and 52 m long and has a total capacity of 1699 persons (including 136 standing places) and a further 190 seats can be placed on the stage. The number of seats can be varied according to the requirements of the performance.



Esztergom Basilica

The Esztergom Basilica is the largest church in Hungary. It is an ecclesiastic basilica in Esztergom, Hungary, the main church of the Archdiocese of Esztergom-Budapest, and the seat of the Catholic Church in Hungary.

It is dedicated for the Blessed Virgin Mary Taken into Heaven, and Saint Adalbert. Its inner area is 56,000 m², 118 m long and 49 m wide. It has a

reverberation time of more than 9 seconds (above 12 seconds at certain frequencies). Its dome, forming a semi-sphere, is situated in the middle, is 71.5 m high inside, with a diameter of 33.5 metres, and is 100 m high from outside, counted from the crypt, thus it is the tallest building in Hungary.



Fertőrákos Cave Theater

Once limestone was quarried in this unique World Heritage site, but now at the cave theatre, opera performances and concerts are held in every summer. Performances have been held regularly since 1970. The excellent acoustics and the wonderful natural surroundings are suitable for arranging musical, dance and opera performances and offers a unique experience. The cave theatre has a seating capacity of 743 persons.



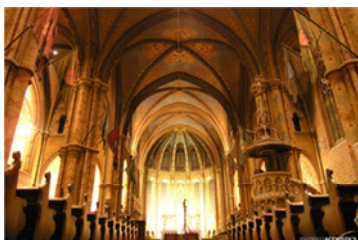
*Liszt Academy – Dome Hall
(before its renovation)*

The Dome Hall of the Liszt Ferenc Academy of music is located on the top floor of the building. The hall accommodates a pipe organ, where various lessons and examinations take place.



*Liszt Academy – Great Hall
(before its renovation)*

The Liszt Ferenc Academy of Music was founded in 1875, under the direction of Ferenc Erkel (who also composed the Hungarian National Anthem) and presidency of Ferenc Liszt himself. The Academy looks back to a 130 year old history, and it is one of the leading musical educational institutes of the world - artists and masters such as Béla Bartók, Zoltán Kodály, Ernő Dohnányi and many more studied and taught here.



*Mátyás Church
(before its renovation)*

At the very heart of Buda's Castle District is the Mátyás-templom (Matthias Church). Officially named as the Church of Our Lady, it has been popularly named after King Matthias Corvinus. In many respects, the 700 year history of the church serves as a symbol of the city's rich, yet often tragic history. Today, Frigyes Schulek's restoration of the church provides visitors with one of the most prominent and characteristic features of Budapest's cityscape. In 1987 the Buda Castle Quarter, including the Matthias Church was honored with the status of World Heritage by the UNESCO World Heritage Committee.

The legendary Matthias Church Choir and Orchestra of the cathedral was celebrating its 320th anniversary in 2008.



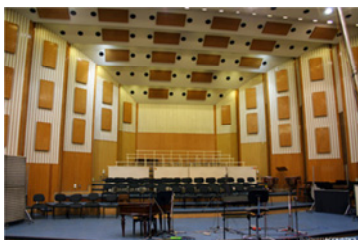
Notre Dame of Kispest

The first people settled in Kispest (currently the 19th district of Budapest) in the second half of the 19th century. Their first church was nothing more but a small chapel with a belfry and a wooden cross. The votive church was built later, in memory of crown prince Rudolf Habsburg who died in 1889. The foundation stone was laid on June 7, 1903 and following a fast construction, the church was consecrated on October 23, 1904.

The brick-covered, 50-meter long and 20-meter wide parish church of neogothic style was designed by Antal Hoffhauser. The tower of the church is a very impressive sight and the church is still a very characteristic mark of the district.

Inside you can find a neogothic, aisleless church, which is 32 meters long (without the altar) and 14 meters wide. The benches of various styles can seat 250-300 people.

The church was renovated between 1998 and 2002, so now, a hundred years later, it can again be seen in its full splendor.

*Studio 22 of the Hungarian Radio*

Studio 22 is one of the most frequently used scoring stages of the Hungarian Radio. This studio, with its clear acoustics, is a perfect place for modern applications: recording classical music or movie scores, from a full orchestra with choir to chamber music. Many celebrities, European composers, world-class musicians and performers recorded in this studio.

*Szeged Cathedral*

The great Szeged Cathedral (also known as the Szeged Votive Church or as the Szeged Dome) has a reverberation time of more than 7 seconds. Having pristine acoustics, it is ideal for orchestral, choir and organ performances. Located near the river Tisza in the southern part of Hungary, the cathedral is the biggest in the region: 66 m (216 feet) in length and 33 m (108 feet) in height, with a capacity of approximately 5000 people.

*Vác Cathedral*

Finished in 1772, the Vác Cathedral stands in the Konstantine square of Vác, Hungary. The mostly classicistic style cathedral is facing the Danube river. Inside the cathedral we find a very bright space which broadens from the choir to the altar and then shrinks back again. Its inner length is 60 m with a 28 m width and 50 m height. The furniture and the frescos are late baroque style, and the choir is sitting on six chorintian pillars. The outer length of the cathedral is 72 m, its width is 34 m, and its dome is 55 m high. Approximately 5000 people can attend a mass and there are 278 seats, and the towers are 38 m high.

3.2 Stereo presets and naming

The default presets contained in the IIL Vienna Edition are the stereo presets. They were recorded by placing a single sound source and a multichannel microphone system in the room.

The naming of these presets contains four identifiers, which for example:

esz_2_A_0

means that 'esz' is from the Esztergom Basilica, '2' is the source position label 'A' is the microphone position label and '0' or '180' are the directions compared to the source. '0' means a direction facing the sources and '180' means a microphone setup that was looking at the opposite direction as the sound source.

Tip: Presets that have a '180' tag in their name will produce a more diffuse, less defined sound with more reverberance as opposed to the '0' tag.

3.3 True stereo presets and naming

True stereo refers to an acoustic setup with two independent monophonic sound sources sounding the same time in the room and one stereo listener capturing their sound (so the term 'stereo' refers to the listener, not the source). True stereo presets can be effectively used to represent a spatially large sound source, as opposed to the default stereo presets which use a single monophonic point source in a room captured with a stereo listener.

The 'true stereo' presets capture two source placements (S1, S2) sounding together and translating to a single stereo listening channel (M1, M2) with the proper crosstalk also taken into account in the reverberation. So, specifically, S1-M1, S1-M2, S2-M1 and S2-M2 are the four internal channels of a true stereo reverberation setup.

The naming of these presets are based on two sources and one microphone. For example:

esz_2_A_180_esz_4_A_180

means that in this preset the source positions labeled as '2' and '4' were captured sounding together at the microphone position 'A', facing 180 degrees (backwards) the sources.

Tip: True stereo setups can be used to simulate a large sound source, such as a big instrument, orchestral recording or many instruments playing together. They however put more demand on the CPU to calculate the reverberation.

3.4 Setting up quad reverberation

If you wish a quad reverberation setup (4 listener channels), it is advisable to use the same source position with the '0' and '180' tags in the preset names, since these refer to the same physical source-microphone placements with the listener facing different directions.

For example, it is possible to setup a quad reverberation by using 'esz_10_A_0' and 'esz_10_A_180', since they use the same 10-A placement. '0' can be considered as the front channels (LF) and (RF) and '180' as the rear channels 'LR' and 'RR'.

It is also possible to use two true-stereo presets for a quad setup, for example:

esz_2_A_0_esz_4_A_0 and *esz_2_A_180_esz_4_A_180*

by using two instances of the VSL Convolution Reverb engine in parallel driven with the same dry sound (make sure however to not to serially chain them since it would not deliver correctly sounding results). Once run in parallel, they will together represent an acoustic setup of using source placements '2' and '4' simultaneously and listening setup (microphone placement) 'A'. The front channels will be provided by the 'esz_2_A_0_esz_4_A_0' and the rear channels by the 'esz_2_A_180_esz_4_A_180' impulse files.

4 A Guide to Room Acoustics

This section, designed as an easily readable introduction to the subject of room acoustics, is also intended as an aid to using ILL Vienna Edition, while (hopefully) proving of general interest.

4.1 Acoustics and sound at a glance

Sound is defined as a mechanical disturbance in a medium – such as air, wood or metal – propagating like a wave and perceived by living creatures through the ability of hearing. Humans usually can perceive sounds in the frequency range of approximately 20 Hz to 20000 Hz (20 kHz) but aging and long exposure to high levels of sound generally limits one's hearing acuity by approximately 2.3 kHz, down from 20 kHz, for every decade after age 20. Sound can travel long distances though no 'material' is actually transported. Acoustics and optics have many similarities but in acoustics, the propagation is mechanical; therefore a propagation medium is also needed.

The disturbance mentioned above is a change in the density of air which happens fast enough so that there is no change in temperature or volume due to this change of density (the process is adiabatic). Sound has two basic descriptor properties, sound pressure and particle velocity. Sound what we perceive can be easily seen as a pressure oscillation that is added onto the constant sound pressure, which is due to the fact that the air above us has finite, quantifiable weight. Consequently, sound pressure is measured as the root-mean-square (RMS) value of the oscillation of the sound pressure in Pa, or Pascal.

$$p = \sqrt{\frac{1}{T_2 - T_1} \int_{T_1}^{T_2} p_{\text{momentary}}^2(t) dt} \quad [\text{Pa}]$$

Sound Pressure Level (SPL) is the level that is calculated by the ratio of this RMS sound pressure and of a reference level of $p_{\text{ref}} = 20 \mu\text{Pa}$ (micro-Pascal), the hearing threshold of a sine wave with a frequency of 1 kHz. The Sound Pressure Level is measured in decibels (dB). The dB scale, in general is a ratio expressed by using the logarithm of a given and a reference number. For easier reading, the abbreviation of SPL is usually attached after the dB.

$$L_p = 10 \log_{10} \frac{p^2}{p_{\text{ref}}^2} = 20 \log_{10} \frac{p}{p_{\text{ref}}} \quad [\text{dBSPL}]$$

The propagation speed of sound in air, c_{air} is a function of many properties such as temperature (T) for example, and can be calculated approximately as follows:

$$c_{\text{air}} = 331.3 \cdot \sqrt{1 + \frac{T}{273.16}} \cong 331.3 + 0.606 \cdot T \quad \left[\frac{\text{m}}{\text{s}} \right]$$

thus the speed of sound is different in cold, as compared to warm air.

4.2 Sound in a room

Room acoustics is the study of acoustics, noise and vibration in rooms, either fully- or partly-closed enclosures. When sound propagates in an enclosed space, it is usually reflected from the boundary surfaces. An *echo* is a reflected sound that we hear clearly as separate to the original sound source, while reverberation is something that we hear as a process or overall effect, the result of many echoes occurring closely in time. Humans perceive two sounds as distinct if their arrival times are separated by at least 30 to 40 ms; even then a sound level difference is also required (Haas effect).

Sound behavior in a room is governed by two main equations which come from the laws of energy conservation and Newton's Second Law. These equations contain a few simplifications to assume linearity (usually correct when talking about everyday sound levels). The first equation is called the linear Euler-equation:

$$\rho_0 \frac{\partial \mathbf{v}}{\partial t} + \frac{\partial p}{\partial \mathbf{x}} = 0$$

where ρ_0 denotes the density of air, $\mathbf{v}$ denotes particle velocity (as a vector), p sound pressure level (as a scalar), t time and $\mathbf{x}$ displacement. This equation roughly says that the mass of materials propagating through a unity area in a given time, and the gradient of the sound pressure sums to zero. The second equation is the so-called continuity equation, and resembles the Euler-equation above:

$$\rho_0 \frac{\partial \mathbf{v}}{\partial \mathbf{x}} + \frac{\partial \rho}{\partial t} = 0$$

With the help of these two equations one can find the *wave equation* describing wave propagation, which is as follows for one dimension:

$$\frac{\partial^2 p}{\partial x^2} - \frac{1}{c^2} \frac{\partial^2 p}{\partial t^2} = 0$$

where c denotes the speed of sound as previously. There is an equation that can be formulated from the wave equation, which has the advantage of not containing the time in it, following the assumption that the sound is a pure sine. This equation is called the Helmholtz-equation:

$$\frac{\partial^2 p}{\partial x^2} + \frac{\partial^2 p}{\partial y^2} + \frac{\partial^2 p}{\partial z^2} + k^2 p = 0$$

or in a shorter form

$$(\nabla^2 + k^2)p = 0$$

where $k = \frac{\omega}{c}$

is the ratio of ω the circular or angular frequency of the pure sine and c the speed of sound; k is called the wave number, and ∇^2 denotes so-called Nabla-operator.

4.2.1 Room modes and its effects on pipe organs

This equation is useful because assuming perfect reflections from walls of a rectangular room, one can find a **room's natural resonance frequencies** called room modes. They are located at certain given frequencies of:

$$\omega_{n_x n_y n_z} = c \sqrt{\left(\frac{n_x \cdot \pi}{l_x}\right)^2 + \left(\frac{n_y \cdot \pi}{l_y}\right)^2 + \left(\frac{n_z \cdot \pi}{l_z}\right)^2} \quad \left[\frac{1}{s}\right]$$

by inserting arbitrary integer numbers of n_x, n_y, n_z to the equation. l_x, l_y, l_z mark the dimensions of a the room in meters.

The number of room modes increases with frequency. In a real room, walls produce imperfect reflections, due to damping, so resonances happen in a frequency range rather than a single frequency. The result is a certain frequency where these ranges overlap in such a way that they can no longer be distinguished separately. This is called the modal overlap crossover frequency or Schröder-frequency and it is defined as:

$$f_{\text{Schröder}} = \sqrt{\frac{c^3}{4 \cdot \ln 10} \cdot \frac{T_{60}}{V}} \quad [\text{Hz}]$$

T_{60} denotes the reverberation time of the room, the time in seconds for the sound pressure level to drop by 60dB (equivalent to one millionth of its initial value), while V is the volume of the room in cubic meters.

This suggests that rooms of large size and small reverberation times, such as modern concert halls, may have a quite low Schröder-frequency, while very large rooms with long reverberation times, such as cathedrals may have a higher Schröder-frequency. Assuming a perfectly diffuse sound field free of acoustic problems, a Schröder frequency below 20 Hz would lead us to expect a sound not unduly affected by single room modes, with little likelihood of very large or annoying changes or sound resonances that can be only heard in certain parts of a room.

For example, shoebox-shaped large concert halls have a Schröder frequency usually somewhere between 16 and 25 Hz, meaning that even for pipe organs, these concert halls shall be treated acoustically using a non-modal approach except for the first few notes of the deepest 32' stops. All this goes to answer the common feeling that deeper pedal notes are typically not so loud in concert halls and why they sound completely different when walking in a cathedral from one location to another.

4.3 Reverberation time and its effects on organ music

One of the basic room acoustic parameters used to describe or evaluate a room's acoustic conditions in an objective way is the reverberation time. This is the time required for the steady-state sound energy density, or sound pressure level to decrease from its original level to only one-millionth, or in other words, decrease by 60 dB. The term 'reverberation time' usually denotes a value expressed in a band of frequencies, while the term decay time denotes a single frequency.

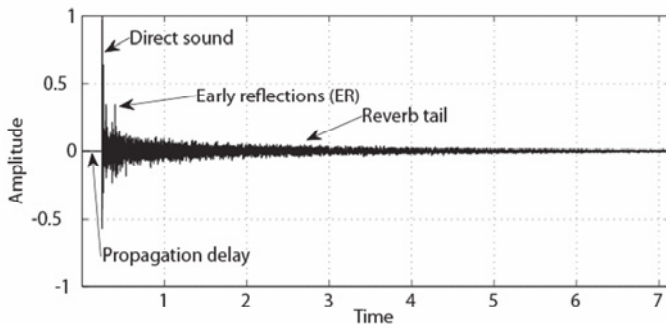
There are different methods and formulas for calculating reverberation time, including Sabine's empirical formula that takes into account room volume, sound absorption, surface and sound speed, but reverberation time can also be empirically measured.

Reverberation can be considered as a finite process that decays exponentially with time in a perfectly diffuse space. In a real room, the value is usually different for different frequencies: lower frequencies have longer, higher frequencies have a shorter reverberation time in most cases. Reverberation at very high frequencies (greater than 8 kHz) in large or absorbent rooms is negligible since air absorbs higher frequency sounds better and in large rooms, such high frequency sounds may not even reach surfaces to reflect from before they are absorbed.

The reverberation time for large concert halls is usually between 1.8 to 2.6 seconds, middle sized churches or smaller cathedrals made of stone usually produce 3 to 4 seconds of overall reverberation while huge cathedrals may have up to 12 seconds or even a bit more.

4.4 The Room Impulse Response

To describe the acoustics of a room objectively, engineers are required to take measurements. There is a nearly 100-year history of room acoustic measurements, but today, these are based on the foundations of System Theory. This theory originally described electrical systems such as circuits, but has now been generalized, providing a good analogy for acoustic and mechanical waves. The system theory helps to find the parameters of an unknown system, which, in this acoustic case, is the room itself. The system has an input and an output, and this is generally all we know about it, along with some assumptions on linearity and time invariance. A signal is input (excite a sound in a room), the output examined (record that sound) permitting the determining of that room's parameters between the measurement points. The signal that holds most of the information within the domain of room acoustics is the room impulse response (RIR).



Room impulse responses (RIRs) are time-domain signals measured in a room during room acoustics measurements. RIRs characterize a room between two given physical locations, a source and a receiver, assuming that the room is a stable, causal, linear and time-invariant (LTI) system. The room impulse response can be imagined easily, it is the response of the room to a short impulsive spike (though, in practice, for several technical reasons signals other than spikes are customarily used and decoded to spikes later).

Room impulse responses have the following parts:

- Propagation delay: time taken for sound to travel from the source to the listener
- Direct sound: in the line of sight, the direct sound is a peak corresponding to the shortest travel path
- Early Reflections (ER) part
 - First reflection: usually the reflection from the ground
 - Second and other reflections: subsequent, still clearly distinguishable reflections
- Reverberation Tail part: this is the stochastic part of the reverberation where so many reflections are present that they can no longer be separately identified.

There is an acoustically important parameter, the Initial Time Delay Gap (ITDG), the time between the direct sound and the first reflection. Sometimes it is defined as the time delay between the direct sound and the first reflection that is not a ground reflection, in the room impulse response. In reverberation software, the term Pre-delay is usually equal to this ITDG parameter.

4.4.1 Other representations of the room impulse response

There are many other – equivalent – representations of room impulse responses in use.

- Time domain
 - **impulse response** (usually abbreviated as IR or RIR)
 - **step response**, obtained by integration of the impulse response
- Frequency-domain
 - **frequency response** function (FRF), obtained by Fourier-transforming the IR
 - Absolute value of the frequency response is the **magnitude response**
 - Angle of the frequency response is the **phase response**
- Complex frequency domain
 - **transfer function** (TF), obtained by taking the Laplace-transform of the IR
- Time-frequency domain (Wigner, Gabor, etc.)
 - Short-term Fourier Transform (STFT) or spectrogram
 - Wavelet transform or scalogram
 - Other time-frequency representations

Usually, the calculation of different room acoustic parameters can be optimized by the use of one or other of these different representations; most of them are used regularly. There are two other forms of time-domain impulse responses used for the calculation of room parameters, but these cannot be transformed back to a RIR once calculated: the Energy Time Curve (ETC) and the Energy Decay Curve (EDC). The ETC is the square of the impulse response and corresponds to the decay of energy (this is often used in reverberation software to display the decay), while the EDC curve is a backward-integrated curve that has the advantage of being completely smooth; determination of the reverberation time of a room is usually made using this curve (and it is often used by scientists).

Once the impulse response of a room has been recorded between several sources and listener positions, it is possible to thoroughly evaluate the acoustic parameters and, if the measurement quality and post processing are of sufficient precision, the room impulse response can also be used to accurately reverberate sound sources. This is called convolution reverberation.

4.5 Understanding convolution reverberation

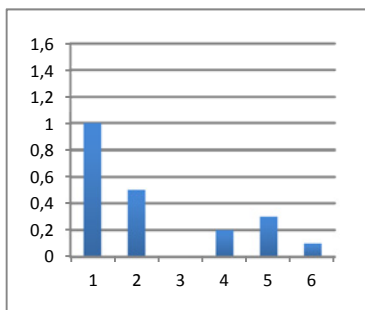
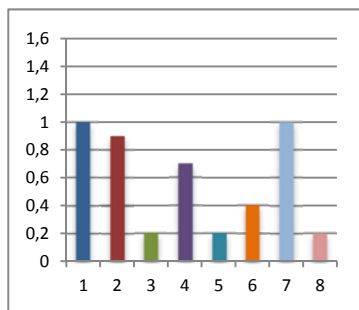
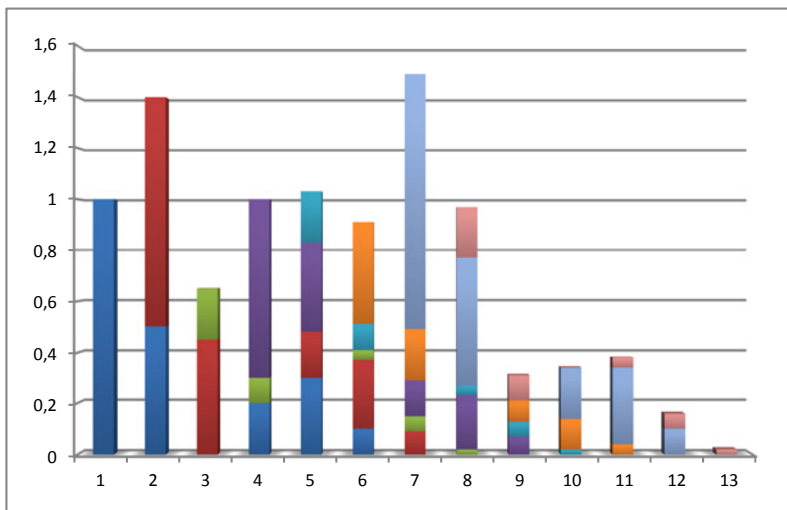
To be able to reverberate a sound (which we will refer to as dry sound from now on), we have to find an algorithm that can calculate the reverberated sound using our original recording and the Room Impulse Response we captured. This algorithm is called convolution. Convolution, in other words, is the way or method to compute the response [result] of a linear time-invariant (LTI) causal system [the acoustical space, e.g. the concert hall] to a known excitation [e.g. our dry sound].

Unlike any other reverberation methods, results of convolution based calculations using impulse responses of real measurements are almost identical (insofar as the equipment and measurement conditions are 'perfect') to what we would indeed experience or record in that very room.

To understand exactly how convolution reverberation works, we will examine the convolution formula (the mathematics), and then visualize all this. We first interpret the sounds with discrete values sampled in time. The sound waves, which are continuous, are converted to discrete amplitude values (quantization) at regular time intervals (sampling rate) by the computer, resulting in a sequence of numbers. A sound source, now in a digitally sampled form, would look like this: (1, 0.9, 0.2, 0.7, 0.2, 0.4, etc.. Let us call this sequence e (excitation) and one of its values as $e[k]$, where k means that we are talking about the k -th sample value. We start counting k from zero, so for example, $e[k = 0]$ or simply $e[0]$ equals to 1 in our case, while $e[1] = 0.9$, etc.. Now let us interpret the Room Impulse Response the same way and call it w (from the name 'weight function'). The reverberated result (the response) would be called y . Now the convolution formula is the summation of the excitation values multiplied by the shifted values of the impulse response:

$$y[k] = \sum_{i=0}^k e[i] \cdot w[k - i]$$

To visualize this, let us look at a sound signal (excitation signal) containing 8 sound samples and an impulse response of 6 samples. Imagine these as short excerpts of a concert hall impulse response and a digitally recorded music, respectively. There are no negative values in these examples for ease of view. The excitation signal is colored differently at each time value so that you can follow the convolution algorithms easily. The horizontal axis is time, the vertical is amplitude. As you can see, each value of the impulse response is multiplied with the source and added together while properly delayed.

**Impulse response****Source signal****Reverberated signal with convolution**

Convolution is very resource consuming for the computer – especially in terms of CPU and memory bandwidth usage. Therefore, as is the case for many other algorithms, optimization methods have been introduced to hasten the convolution process, allowing real-time or near real-time calculation for use in audio applications. The discrete convolution formula above in reality simply 'applies' finite length impulse response (FIR) filter to the input, where the filter coefficients are the samples of its impulse response. The term 'filter' is used for anything that has an impulse response in system theory. Traditional personal computers or servers even nowadays are often not capable of applying very long impulse responses directly as FIR filters, due to the high computational load.

Therefore, optimization is needed, leading to the application of frequency domain convolution; however, transforming a time-domain signal to frequency domain requires all the time-domain samples to exist a-priori, which means a significant latency is introduced, an unavoidable delay caused by collecting the required samples before we can use them for calculations. The transformation to the frequency domain – called the Discrete Fourier Transform (DFT) – can be calculated very fast with its implementation called the Fast Fourier Transform (FFT). Convolution in the frequency domain is simply a multiplication, so after conducting the multiplication, the inverse Fourier Transform (IDFT) is applied to obtain the time-domain signal of the result. Although this is very effective computationally, it cannot be applied successfully to audio in real time because of the latency. A good compromise can be achieved by dividing the incoming signal and the impulse response into parts, applying the processing to these parts separately before combining their outputs. This increases the computational load somewhat, but still is more effective than the direct time-domain convolution and also decreases the latency compared to the direct frequency-domain convolution. This algorithm is usually called 'fast convolution', and is implemented by the partitioned convolution method. Other optimization methods are also known but these are based on dropping away some information in order to make the calculations simpler. Many reverberation software use this 'fast convolution' technique to support low latency reverberation with realistic CPU loads.

4.6 Acoustical parameters

From the room impulse response and other acoustic measurements, a wide range of acoustical parameters can be calculated, which may be classified into the following categories for easier reading, together with some examples:

- Temporal parameters
 - Initial time-delay gap (ITDG, t_1)
 - Early reflections - tail boundary, ER-limit (transition time, t_e)
 - Reverberation time (RT60, T10, T20, T30, etc.)
 - Early decay time (EDT)
 - Bass ratio (BR)
 - Treble ratio (TR)

- Rise time (TR)
- Inversion index (II)
- Steepness (sigma)
- Temporal diffusion (Delta)
- Energetic parameters (energies and energy ratios)
 - Clarity index (C7, C50, C80)
 - Definition (D50, D80)
 - Strength (G, G80, GL)
 - Centre time (Ts)
 - Reverberant to early sound ratio (R)
 - Hallmass, Hallabstand (H)
 - Echo Criterion (EC)
 - Spectral Density (S)
 - Early Energy Balance (EEB)
- Spatial parameters
 - Lateral Strength (GEL, GLL)
 - Inter-Aural Cross-correlation Coefficient (IACC-A, IACC-E, IACC-L, IACC-E3, IACC-L3)
 - Lateral Energy Fraction, Early Lateral Energy Fraction (LF, LFC, ELEF)
 - Lateral Efficiency (LE)
 - Spatial Impression (SI)
 - Subjective Spaciousness (SS)
 - Room Response (RR)
- Stage parameters
 - Support (ST1, ST2, ST-early, ST-late, ST-total)
 - Clarity on Stage (CS)
 - EDT on Podium (EDTP)
 - Early Ensemble Level (EEL)
- Quantitative Speech Intelligibility Parameters
 - Speech Transmission Index (STI, STIr, STIEL, STIPA, RaSTI)
 - Speech Intelligibility Index (SII)
 - Articulation loss of Consonants (AlCons), Privacy Index (PI)
 - Radius of distraction (rd), radius of privacy (rp)

The above mentioned parameters are not fully independent, this means if one changes, others may also change in some relationship. Scientists continue to seek independent parameters, so as to avoid the need for so great a range of interactive parameters. Some already established 'orthogonal parameters' are known as:

- Listening level (LL)

- Initial time delay gap (ITDG; Δt_1); without the ground reflection
- Subsequent reverberation time (T_{sub})
- Inter-aural cross-correlation coefficient (IACC)

4.7 Objective and subjective pairing

The following list shows the subjective acoustic impression paired to its corresponding objective acoustical parameter, using today's standard terminology.

- Reverberance: early decay time (EDT)
- Liveness: EDT, reverberation time (RT)
- Fullness: RT
- Loudness: strength (G)
- Clarity: clarity index for music (C80)
- Timbre: reverberation time as a function of frequency, $RT(f)$
- Warmth: bass ratio (BR)
- Brilliance: treble ratio (TR) or high ratio (HR)
- Stage support: ST1
- Hall response: ST-late
- Spaciousness: lateral fraction (LF), early inter-aural cross correlation coefficient IACC-E
- Apparent source width: LF at (500-4000 Hz), IACC-E
- Listener envelopment: late lateral fraction (LLF) at (125-500 Hz), late IACC (IACC-L)
- Intimacy: LF, IACC-E
- Ensemble: clarity on stage (CS), early decay time on podium (EDTP)

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